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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/388,010	09/01/1999	RUSSELL H. LAMBERT	15-0195	3282

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PATENT COUNSEL
TRW INC
SPACE & ELECTRONICS GROUP
ONE SPACE PARK E2 6072
REDONDO BEACH, CA 90278

EXAMINER	
ARMSTRONG, ANGELA A	
ART UNIT	**PAPER NUMBER**
2654	13

DATE MAILED: 07/17/2003

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)
	09/388,010	LAMBERT ET AL.
	Examiner	**Art Unit**
	Angela A. Armstrong	2654

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
- Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on _01 May 2003_ .

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) _1,4-6,9 and 10_ is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) _1,4-6,9 and 10_ is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

11)☐ The proposed drawing correction filed on _____ is: a)☐ approved b)☐ disapproved by the Examiner.

If approved, corrected drawings are required in reply to this Office action.

12)☐ The oath or declaration is objected to by the Examiner.

Priority under 35 U.S.C. §§ 119 and 120

13)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____ .

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

14)☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. § 119(e) (to a provisional application).

a) ☐ The translation of the foreign language provisional application has been received.

15)☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. §§ 120 and/or 121.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449) Paper No(s) _____ .
4) ☐ Interview Summary (PTO-413) Paper No(s). _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: .

DETAILED ACTION

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 1, 4-6, and 9-10 are rejected under 35 U.S.C. 103(a) as being unpatentable over Marash (US Patent No. 5,825,898) in view of Coker (US Patent No. 4,581,758).

3. Marash teaches a system and method for adaptive interference canceling for reducing interference in a signal received from an array of sensors.

4. Regarding claims 1 and 6, at col. 4, lines 50-67 Marash teaches a sensor array having individual sensors which receives signals from a signal source and from interference sources (which reads on a plurality of microphones positioned to detect speech from a single speech source and noise from multiple sources), producing a main channel representing signals received in the direction of the source, such that the main channel contains both a source signal component and interference signal component (which reads on generating microphone output signals, such that the reference microphone receives acoustic signals both from the speech source and from the multiple noise sources) producing a reference channel representing signals from directions other than the that of the signal source.

Marash does not specifically teach a plurality of bandpass filters for eliminating spectral bands containing noise from the microphone output signals. Coker et al teach a system of acoustic direction identification of a desired sound source in a noisy reverberant environment.

Specifically Coker et al teach implementation of bandpass filters for removing low-frequency components of speech and for eliminating high frequency noise, which can produce unwanted spurious events (col. 3, lines 34-41).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the adaptive interference canceling system of Marash to implement bandpass filtering of the input signal as suggested by Coker et al, for the purpose of eliminating high frequency noise which can produce unwanted spurious events, as taught by Coker et al.

Although applicant has included the limitation of "without the need for noise estimation techniques…", the described limitations of the adaptive filters and the signal summation circuit are the delay-and-sum beamforming and noise estimation process.

At col. 8, lines 23-67, Marash teaches a plurality of adaptive filters to process the output signals from the reference channels with the output signals from the main channel, which reads on the plurality of adaptive filters, for aligning each data microphone output signal with the output signal from the reference microphone.

At col. 6, lines 1-27, Marash teaches the processing of the main channel matrix to produce a main channel as a weighted sum of outputs which filters a signal coming in all directions to produce a signal coming in a specific direction, which reads on the signal summation circuit for combining filtered output signals from the microphones whereby signal components resulting from speech are combined.

At col. 6, lines 28-48, Marash teaches the processing of the reference channel matrix to produce a reference channel as a weighted sum of interference signals, which reads on a signal

summation circuit for combining filtered output signals from the microphones whereby signal components resulting from noise are combined.

At col. 8, lines 46-48, Marash discloses a difference unit for subtracting the reference channel matrix (interference channel) from the main channel matrix (the source and interference channel) to effectively reduce the interference in the signal of interest, which reads on the speech conditioning circuitry coupled to the signal summation circuit to reduce reverberation effects.

Neither Marash nor Coker teaches speech detection and enabling the adaptation process only when speech is detected. However, it would have been obvious to one of ordinary skill at the time of the invention to modify the adaptive interference reduction system of Marash and Coker to implement enabling the adaptation process only when speech is detected, because such a modification would eliminate unnecessary filter weighting adaptation processing, making the system more efficient and economical.

5. Regarding claims 4-5 and 9-10, at col. 8, line 23 continuing to col. 9, line 40, Marash discloses a frequency selective constraint adaptive filter which implements a Least Mean Square (LMS) algorithm for adaptive filtering via updating or adjusting the filter weights. Specifically at col. 8, line 27-35 Marash teaches that the outputs of the reference channel passes through the a FIR filter of adjusted filter weights, which reads on the filtering data microphone output signals by convolution with a vector of weight values. At col. 8, lines 47-48 and col. 9, lines 2-3, Marash discloses the calculation of an error signal e (n) that is the difference between the main channel signal and the canceling signal, which reads on comparing the filtered data microphone output signals with reference microphone output signals and deriving an error signal. At col. 8, lines 59-62, Marash discloses that the invention utilizes the LMS algorithm which minimize the

difference between the main channel and the canceling signal, which reads on adjusting the weight values convolved with the data microphone output signals to minimize the error. At col. 8, lines 51-54, Marash teaches that when the filter weights settle, the output signal generates a source signal substantially free of the actual interference signal, which reads on the repeating the filtering, comparing and adjusting steps to converge on a set of weight values that results in minimization of noise effects. At col. 9, lines 6-40, Marash discloses the frequency-selective weight constraint unit which after the filter weights have been adapted to minimize the error, a FFT receives the adaptive filter weights and performs an FFT of the weights, truncates values of the filter weight representation and converts the weights with an IFFT, which reads on the transforming of the filter weights to the time domain, zeroing out portions of the filter weight values and converting the filter values back to a frequency domain.

Response to Arguments

6. Applicant's arguments filed May 1, 2003 have been fully considered but they are not persuasive.

Applicant argues there is no suggestion in Marash or Coker to include speech detection to enable adaptive filtering. In response to applicant's argument that there is no suggestion to combine the references, the examiner recognizes that obviousness can only be established by combining or modifying the teachings of the prior art to produce the claimed invention where there is some teaching, suggestion, or motivation to do so found either in the references themselves or in the knowledge generally available to one of ordinary skill in the art. See *In re Fine*, 837 F.2d 1071, 5 USPQ2d 1596 (Fed. Cir. 1988)and *In re Jones*, 958 F.2d 347, 21

USPQ2d 1941 (Fed. Cir. 1992). In this case, one of ordinary skill would have the knowledge to modify the adaptive interference reduction system of Marash and Coker to implement enabling the adaptation process only when speech is detected, because such a modification would eliminate unnecessary filter weighting adaptation processing, making the system more efficient and economical.

Applicant argues that the main channel matrix of Marash has nothing to do with reducing reverberation effects. The Examiner disagrees and argues at col. 8, lines 46-48, Marash discloses a difference unit for subtracting the reference channel matrix (interference channel) from the main channel matrix (the source and interference channel) to effectively reduce the interference in the signal of interest, which reads on the speech conditioning circuitry coupled to the signal summation circuit to reduce reverberation effects, since reverberation effects is a from of signal interference.

Applicant argues there is nothing in either Marash or Coker to suggest that the bandpass filters of Coker might be advantageously combined with the system of Marash. In response to applicant's argument that there is no suggestion to combine the references, the examiner recognizes that obviousness can only be established by combining or modifying the teachings of the prior art to produce the claimed invention where there is some teaching, suggestion, or motivation to do so found either in the references themselves or in the knowledge generally available to one of ordinary skill in the art. See *In re Fine*, 837 F.2d 1071, 5 USPQ2d 1596 (Fed. Cir. 1988)and *In re Jones*, 958 F.2d 347, 21 USPQ2d 1941 (Fed. Cir. 1992). In this case, Coker specifically teaches that implementation of bandpass filters is advantageous for removing low-frequency components of speech and for eliminating high frequency noise, which can produce

unwanted spurious events (col. 3, lines 34-41). Thus, one of ordinary skill would be motivated to modify the adaptive interference canceling system of Marash to implement bandpass filtering of the input signal as suggested by Coker et al, for the purpose of eliminating high frequency noise which can produce unwanted spurious events, as taught by Coker et al.

7. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Angela A. Armstrong whose telephone number is 703-308-6258. The examiner can normally be reached on Monday-Thursday 7:30-5:00 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (703) 305-9645. The fax phone numbers for the organization where this application or proceeding is assigned are 703-872-9314 for regular communications and 703-872-9314 for After Final communications.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the receptionist whose telephone number is 703-306-0377.

Angela A. Armstrong
Examiner
Art Unit 2654

AAA
July 13, 2003

Richemond Dorvil
Primary Examiner